IN THE CLAIMS:

1. (Currently amended) Array microphone with several individual microphones (1-4) connected with a signal processor (11) that comprises at least one digital filter for each individual microphone, in particular for voice recognition, characterized in that at least one loudspeaker *(5)* is provided, which is arranged in the acquisition ~~area~~ range of each of the individual microphones (1-4), that an electronic circuit is provided, which applies a signal to the loudspeaker *(5)* in such a manner that it emits a predetermined periodic noise signal, and that the signal processor (11) evaluates the response signals coming from each of the microphones and/or from each of the digital filters as a response to the reception of the periodic noise signal.

2. (Currently amended) Method for checking array microphones, comprising several individual microphones (1-4) connected with a signal processor (11), that comprises at least one digital filter for each individual microphone, characterized in that at least one loudspeaker *(5)* is provided in the acquisition ~~area~~ range of each of the individual microphones (1-4) and connected with a signal processor (11), to which each microphone **(1-4)** is also connected, and that the signal processor (11) emits a predetermined periodic noise signal via the

loudspeaker *(5)*, that the signal processor (11) evaluates the response signals that subsequently come from each individual microphone **(1-4)** and/or from each of the digital filters, and compares them with model signals stored in the signal processor (11) or externally, and which correspond to properly operating individual microphones (1-4) or properly operating filters, and that the signal processor (11) provides a display in the form of a message and/or stores the deviation of the response signals from the model signals.

3. (Original) Method according to Claim 2, characterized in that the signal processor (11), before emitting a predetermined periodic noise signal via the loudspeaker *(5)*, carries out a verification of the loudspeaker *(5)*, where the loudspeaker signal is directly applied to the AID converter (9) and said loudspeaker is operated in parallel to the input impedance of the AID converter (9), and where the loudspeaker *(5)*, together with the output resistance of the output amplifier (7) which operates the loudspeaker *(5)*, forms a voltage divider, and that the signal applied to the A/D converter (9) is recorded and evaluated by comparing this signal with a reference signal that originates from the measurement with a reference impedance instead of the loudspeaker impedance.

4. (Original) Method according to Claim 3, characterized

in that the ratio of the loudspeaker impedance to the input impedance of the AID converter (9) is verified and, if it deviates too far from the value of 1, is adjusted by an additional pre-resistance, which is switched in front of the loudspeaker *(5)*.

5. (Currently amended) Method for the automatic calibration of array microphones, comprising several individual microphones (1-4) connected to a signal processor (11) that comprises at least one digital filter for each individual microphone, whereby the signal processor (11) increases the ~~bundling degree~~ sound power concentration of the array microphone and suppresses lateral sound sources by means of an appropriate algorithm applied to the individual microphone signals, whereby filter coefficient sets used in the digital filters and which are characteristic for the arrangement, type, sensitivity, and characteristics of the used individual microphones (1-4), the acoustical environment, and the location of the sound sources are components of the algorithm, characterized in that at least one loudspeaker *(5)* is provided in the acquisition ~~area~~ range of each individual microphone (1-4), which loudspeaker is connected with a signal processor (11), to which each individual microphone (1-4) is also connected, in that the signal processor (11) emits via the loudspeaker *(5)*, a predetermined periodic noise signal, that the signal processor (11) evaluates the response signals that

subsequently come from each individual microphone (1-4) and/or from each digital filter and compares them with model signals which are stored in the signal processor (11), or externally, and which correspond to properly operating individual microphones (1-4) or properly operating digital filters, and that the signal processor (11), as a function of the deviation of the response signals from the model signals, changes the value of individual filter coefficients or of all the filter coefficients of the filter coefficient set and repeats the test until the response signals are in the range of the model signals.

6. (Original) Method according to Claim 5, characterized in that, after a predetermined number of test repetitions have been carried out, the test is interrupted and an error message is displayed and/or stored.